Timing Recovery for the Magnetic Recording Channel Using the Wave Difference Method

John P. Keane and Paul J. Hurst

***Abstract*— Timing recovery using the wave difference method (WDM) is proposed for use in a magnetic recording read channel. The scheme uses an oversampling ratio of two and is suitable for mixed-signal or digital implementations. A simple class of prefilter that significantly improves the signal to interference ratio (SIR) of the generated timing-error estimate is described. This prefilter also allows a simple implementation based on an absolute value, rather than squaring or fourth-order nonlinearity. Simulation results for this scheme are presented for a range of recording densities and channel noise levels.**

***Index Terms*— Magnetic recording, Timing circuits, Jitter.**

I. INTRODUCTION

MANY common timing recovery schemes [1] are based on an inductive architecture where the sampler is contained within a feedback loop. Decision-directed inductive schemes [2], where output decisions are used to aid the timing recovery, are often used for the magnetic recording channel as they improve performance in the presence of high levels of channel distortion. However, as the detector is inside the timing recovery loop, unwanted interactions can occur, especially when the detector is adaptive.

Some timing recovery schemes instead use a deductive architecture where the sampling clock is generated from a timing tone extracted from the input signal before the sampler. Such schemes have the advantage of being independent from the sampler and the detector. Timing recovery schemes using this approach have been proposed for the magnetic recording channel, where performance may be improved when channel SNR is low [3]. However, these schemes used continuous-time filters which are difficult to implement in a VLSI process.

This paper describes a discrete-time approach based on the wave difference method [4]. This non-data-aided inductive scheme recovers the sampling clock directly from samples of the input signal. It shares the advantage of independence from the detector with the deductive scheme described above and is simpler to implement as it operates in discrete time.

II. PROPOSED ARCHITECTURE

Much previous work [3], [5]–[7] presents timing recovery based on timing tone extraction using analog continuous-time signal processing. A typical architecture is illustrated in Fig. 1. The input signal $x(t)$ is passed through an optional prefilter $C(\omega)$ to produce $f(t)$ before being acted on by a memoryless nonlinear operation $w(.)$. Common nonlinearities used include squaring [3], fourth-order [6] and absolute value [7] functions. The nonlinearity generates the signal $s(t)$ containing a timing tone at frequency $\omega_S = 2\pi/T$ where T is 1 bit period. A phase locked loop (PLL) matches the phase and frequency of the local oscillator to this tone. This architecture contains analog continuous-time filters. Accurately tuned filters can be difficult to implement in a VLSI process, so a discrete-time approach that avoids continuous-time filtering is preferable.

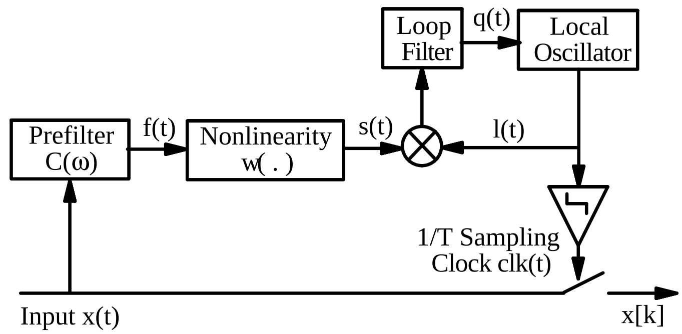

Fig. 1. Continuous time architecture using a PLL.

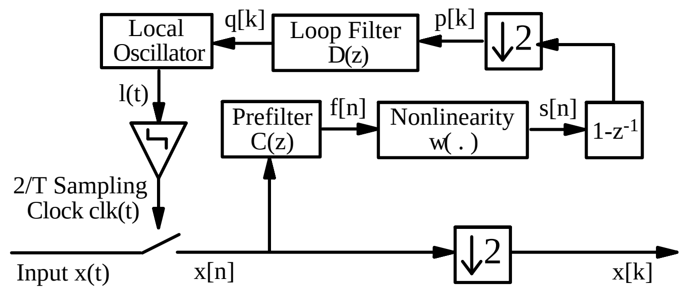

Fig. 2. Discrete-time architecture using the wave difference method (WDM).

An example of discrete-time timing recovery based on nonlinear timing tone extraction is the wave difference method (WDM) [4], which has previously been proposed for use in a digital subscriber loop. The proposed architecture using this approach is shown in Fig. 2.

The input signal $x(t)$ is sampled at a rate $2/T$. The samples $x[n]$ are passed through an optional prefilter $C(z)$ and are then acted on by a memoryless nonlinearity $w(.)$ to generate a timing tone at frequency ω_S in $s[n]$. A first-order differencing operation $(1 - z^{-1})$ removes the DC component of $s[n]$ before down-sampling by 2. Down-sampling causes the tone at ω_S to be aliased to DC producing the timing error estimate $p[k]$.

The fact that $p[k]$ gives a useful timing error estimate can be understood intuitively as follows. If we remove the prefilter $C(z)$, then, since the nonlinearity $w(.)$ is memoryless, the samples $s[n]$ can be considered as samples of a continuous-

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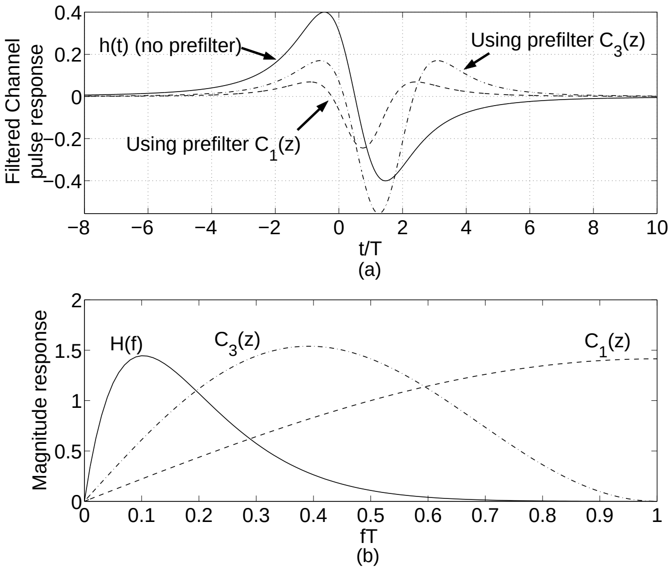

Fig. 3. (a) Effective channel pulse response for D=3 before and after prefiltering (b) Magnitude responses of channel $H(f)$ and prefilters

time signal $s(t)$ taken at times $\tau + kT$ and $\tau + kT + \frac{T}{2}$, where $s(t)$ is the result of the nonlinear operation $w(.)$ being applied to the input signal $x(t)$. If $s(t)$ contains only DC and a timing tone at ω_S, then there exists a timing offset τ_0 such that $s(\tau_0 + kT) = s(\tau_0 + kT + \frac{T}{2})$. This corresponds to sampling $s(t)$ at the zero crossings of the timing tone. Consider

$$p[k] = s(\tau + kT) - s(\tau + kT + T/2). \tag{1}$$

For τ close to τ_0, $p[k] \propto (\tau - \tau_0)$, so we can use $p[k]$ as a measure of the timing error in order to converge to τ_0.

In disk drives, a detector operating on $x[k]$ often requires a timing offset at a multiple of $\tau = T/2$ for optimum operation [1]. However, the timing recovery loop in Fig. 2 converges to $\tau = T/4$ when no prefilter is used. One solution [4] is to precede the down-sampling of $x[n]$ in Fig. 2 by an interpolating filter with a delay of an odd multiple of $T/4$.

III. Prefiltering

The signal-to-interference ratio (SIR) of the tone at ω_S in $s[n]$ determines the jitter of the recovered clock $clk(t)$. The SIR is defined as the ratio of the power of the tone at ω_S to the variance of $s[n]$ when the input is sampled at a time corresponding to the zero-crossings of this tone. This interference is signal-dependent and determined by the channel pulse response. In particular, a certain class of pulse response was found that exhibits virtually no interference at the zero crossings of the tone produced by the nonlinearity when using the continuous-time architecture in Fig. 1 [5]. Appropriate prefiltering [5], [6] of the signal before the nonlinearity can shape the channel response to closely approximate such an ideal pulse response and significantly reduce the resulting clock jitter.

Applying this technique to the discrete-time case, a discrete-time prefilter was added to the WDM architecture described in [4] to change the effective pulse response of the signal applied to the nonlinearity.

In the magnetic recording channel, the transition response $g(t)$ is modeled by a Lorentzian pulse with a 50% pulse width (PW_{50}) of DT, where D is referred to as the recording density [1]. Hence the pulse (dibit) response of the channel, as shown in Fig. 3(a), is modeled by the Lorentzian dibit response

$$h(t) = g(t) - g(t - T) \tag{2}$$

This channel frequency response $H(f)$ as shown in Figure 3(b) contains little high-frequency information, so high-pass prefiltering is required to make it match more closely the ideal class of pulse responses in [5].

Several additional criteria were considered when designing the prefilter $C(z)$ in Fig. 2. $C(z)$ should have:

1) a constant group delay that is an odd multiple of $T/4$, so the timing recovery converges to the optimum sampling phase, removing the need for an interpolating filter before the baud rate detector; and
2) a low frequency phase shift of $\pi/2$, to convert the odd pulse response $h(t)$ to an even pulse response. Since the WDM relies on the symmetry of the pulse response, converting from an asymmetric odd response to a symmetric even response increases the SIR of the resulting timing error estimate.

A class of filters that satisfies all of these criteria is a FIR filter $C(z)$ of odd order N with

$$c[n] = -c[N - n] \tag{3}$$

Such a filter, operating at a $2/T$ sample rate, has a group delay of $NT/4$ and a phase shift of $\pi/2$ at DC. Two such filters are considered here. A first-order difference filter with transfer function

$$C_1(z) = \left(1/\sqrt{2}\right)\left(1 - z^{-1}\right) \tag{4}$$

and a third-order filter with transfer function

$$C_3(z) = 0.5\left(1 + z^{-1} - z^{-2} - z^{-3}\right) \tag{5}$$

The magnitude responses of these filters and the resulting effective pulse responses for a magnetic recording channel with D=3 are plotted in Fig. 3. The effective pulse response results from applying the discrete-time operation provided by the prefilter to the channel response $h(t)$. In Fig. 3(b) the prefilter $C_1(z)$ is seen to provide boost at high frequencies, well beyond the signal band, which may degrade the SNR due to noise enhancement. Higher order filters could also be used to reduce interference further, but they increase the implementation complexity and add undesired delay to the timing recovery loop.

IV. Simulation Results

In order to estimate the quality of the timing error estimate generated using this method, simulations were performed on the system in Fig. 2 with the feedback loop broken so that the effect of phase errors on $p[k]$ can be measured. The channel was modeled by a Lorentzian dibit response with additive white Gaussian noise. As the sampling phase is varied, the mean value of the timing error estimate $p[k]$ describes a sinusoid. The samples $p[k]$ were passed through a 1-pole

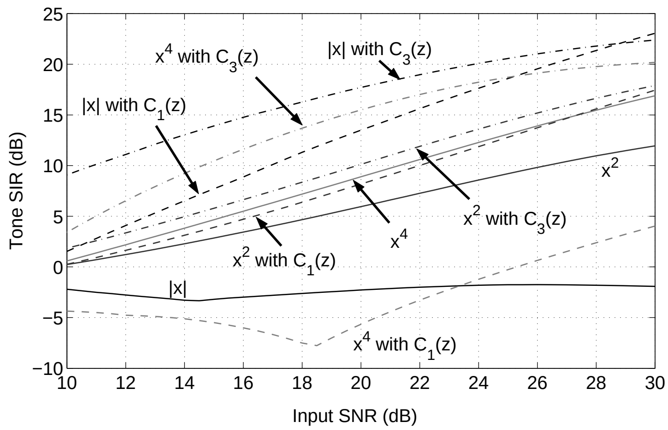

Fig. 4. SIR at zero crossings with D=3.

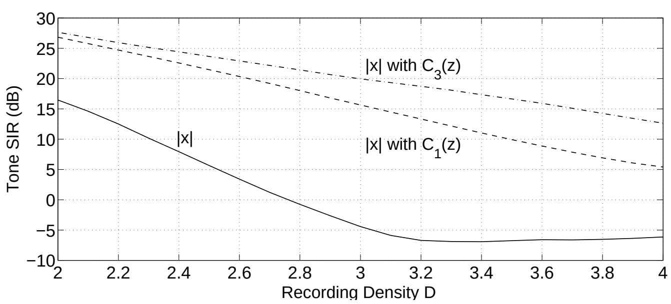

Fig. 5. SIR at zero crossings with input SNR=18dB and absolute value nonlinearity.

low-pass filter with bandwidth 0.1% of ω_S. This models the filtering action of the timing recovery loop.

The SIR at the zero crossings of the output of this filter, which determines the jitter in the recovered clock, is plotted in Fig. 4 for $PW_{50} = 3T$ as the input SNR is varied. Results are plotted for three different nonlinearities both with and without prefiltering. Prefiltering with $C_1(z)$ provides high frequency boost. This boost increases the tone power using the square nonlinearity as the tone power in this case is dependent on signal power around $f = 1/2T$. Prefiltering with $C_1(z)$ provides no benefit for the fourth-order nonlinearity as it attenuates channel frequency components around $f = 1/4T$, which contribute to the tone power in this case. Prefiltering with $C_3(z)$ improves the tone SIR for both square and fourth order nonlinearities. Although the equivalent pulse response in Fig. 3(a) has a similar shape for both prefilters, $C_3(z)$ provides less high-frequency noise boost than $C_1(z)$ and so gives a higher SIR at low input SNR. For low input SNR, channel noise limits the SIR of $p[k]$. With each prefilter, the absolute value method produces the highest SIR for all input SNRs.

Fig. 5 shows the SIR performance using this method for an input SNR of 18dB when the recording density D is varied. With a prefilter, the SIR performance decreases as D increases. This decrease is smooth however, so small variations in recording density do not abruptly change performance.

In order to evaluate the jitter performance of the closed-loop architecture in Fig. 2, simulations were performed using a first-order loop filter with coefficients resulting in a loop bandwidth of approximately 0.1% of ω_S. The resulting rms absolute jitter (i.e. the rms deviation of the sampling time about the optimum timing phase) is plotted in Fig. 6 for the absolute value nonlinearity using both $C_1(z)$ and $C_3(z)$ prefilters. Simulation results for a simple decision-directed feedback timing recovery scheme [8], are shown for comparison. This is a version of the approach in [2] adapted for use with a noise-predictive decision-feedback equalizer that was shown in [8] to allow near-optimal performance of this detector. It can be seen that the timing recovery method described here with prefilter $C_3(z)$ achieves comparable jitter performance to this decision-directed feedback scheme.

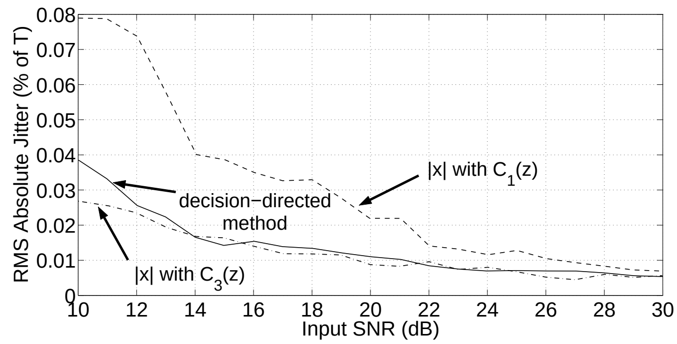

Fig. 6. RMS absolute jitter using the absolute value with prefiltering compared to a simple decision-directed scheme.

Conclusion

A timing recovery scheme for the magnetic recording channel that operates independently of the detector has been described. The addition of a prefilter has been shown to dramatically improve the performance obtained using this approach and allows the use of the absolute value nonlinearity, which is simpler to implement than a square law or fourth order nonlinearity. Simulation results verify that this method achieves a jitter level that is comparable to that of a simple decision-directed scheme.

References

[1] J.W.M. Bergmans, *Digital Baseband Transmission and Recording*, Kluwer Academic Publishers, 1996.

[2] K. H. Mueller and M. Müller. "Timing recovery in digital synchronous data receivers," *IEEE Trans. Commun.*, vol. 24, pp. 516-531, May 1976.

[3] S. Raghavan and H.K. Thapar, "Feed-forward timing recovery for digital magnetic recording," in *Proc. Int. Conf. Commun.*, 1991, pp. 794-498.

[4] O. Agazzi, C.-P.J. Tzeng, D.G. Messerschmitt and D.A. Hodges, "Timing recovery in digital subscriber loops," *IEEE Trans.Commun.*, vol. 33, pp. 558-569, June 1985.

[5] A.N. D'Andrea, U. Mengali and M. Moro, "Nearly optimum prefiltering in clock recovery," *IEEE Trans. Commun.*, vol. 34, pp. 1081-1088, Nov. 1986.

[6] T. Fang and C.-F. Liu, "Fourth-power law clock recovery with prefiltering," in *Proc. Int. Conf. Commun.*, 1993, pp. 811-15.

[7] C.M. Melas and H.K. Thapar, "On timing recovery from full-wave rectified magnetic signals," *IEEE Trans. Magn.*, vol. 34, pp. 91-93, Jan. 1998.

[8] J.P. Keane, M.Q. Le and P.J. Hurst, "Analog timing recovery for a noise-predictive DFE," *IEEE J. Solid-State Circuits*, vol. 38, pp. 338-342, Feb. 2003.